Brass Tacks

An in-depth look at a radio-related topic

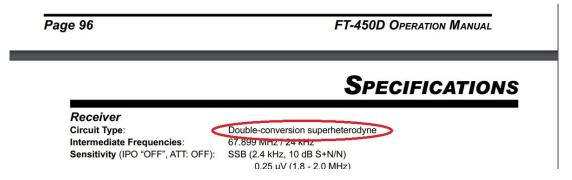






Direct sampling

When looking at the specs of a transceiver under the *Receiver* section, you might have noticed a designation such as *Triple-conversion superheterodyne*, *Double-conversion superheterodyne*, or *Direct Sampling*. Some even feature a hybrid, called *Direct Sampling Superheterodyne*. For most of us, those terms mean little. But to those who are concerned with technological advancements in amateur radio, these could represent steps in our progress toward better receiver performance.



Double-conversion superheterodyne receiver specification of the Yaesu FT-450D

As we've moved into the Digital Age, more and more activities have made the transition from purely analog operations to binary, and amateur radio is no exception. In a previous article, I explained what transmitted digital signals look like when viewed on an oscilloscope, for example. This time, let's explore what it takes to convert a received signal to a digital format, and why we would even want to do such a thing.

Receiver Receiver system Direct Sampling Intermediate frequency Sensitivity* SSR/CW (at 10dR S/N)

Direct Sampling receiver specification of the Icom IC-7300

Nearly all amateur radio transceivers manufactured today are designed with some amount of digital circuitry, due to the lower cost associated with integrated circuits rather than discrete components (fewer components that can fail), the physical savings in size and weight they present, and the increased ability to process a signal digitally over that of analog methods. This digital heavy lifting is provided primarily by two sections: the ADC (analog-to-digital converter) and the DSP (digital signal processor).

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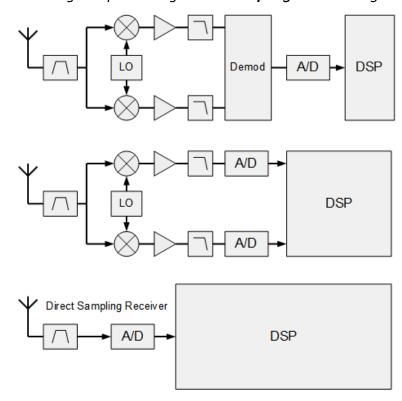
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As you can see from the following evolutionary diagram, more than twenty years ago, amateur radio transceivers began incorporating DSP, but the large part of the incoming signal's processing still required down-conversion (changing to a lower, easier-to-handle frequency) and demodulation (extracting the audio or other desired signal.) Today's newest transceivers no longer need to down-convert an incoming signal, due to the higher conversion speed of the ADC (labeled here as "A/D"), and demodulation is completely encapsulated in the DSP, essentially turning the receiver into an *SDR* (software-defined radio.) Because the ADC can "sample" the incoming signal directly (that is, without mixing or any prior processing), we refer to that method of signal handling and processing *Direct Sampling* or *Direct Digital Conversion*.



The job of the DSP, then, is to demodulate the signal, and remove as much noise from the final output signal as technologically possible. And today's digital enhancements, combined with advanced discrete mathematics, are able to refine the signal to a much greater level than has ever been previously achieved. It's beyond the scope of this article to explore the details of modern DSP, but many have asked how an analog signal can be transformed into a digital signal, so let's focus on how an ADC works.

At very consistent intervals, the ADC will take "samples" of the analog signal, by simply measuring its amplitude at those times. Those moments and their intervals are collectively set by a "clock" signal, which operates at a very high frequency, constantly changing from a high level to a low level, then back again. Essentially, the ADC is acting like a voltmeter, recording a list

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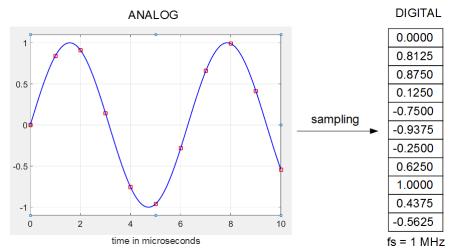




age values, which vary between a negative amount to a positive amount, are then converted into a list of binary values that are usable by the DSP.

For example, if the lowest incoming voltage value (after some amplification) is -3.7 volts, and

For example, if the lowest incoming voltage value (after some amplification) is -3.7 volts, and the highest is +3.7 volts, the ADC might assign the binary value 00000000 to -3.7 volts and 11111111 to +3.7 volts, for a possible 256 voltage value representations. To remove the need of handling negative numbers, a "bias" of 3.7 is added to each value to *normalize* it. The value is then converted to binary by dividing the old range (-3.7 to +3.7, or +3.7 -3.7 = 7.4) and multiplying the new range (0 to 256, in this case, or 256 - 0 = 256.) This way, when the ADC detects +1.1 volts at one point, it'll record $(1.1 + 3.7)/(7.4) \times 256 = 166 = 0 \times A6 = 10100110$ as its binary representation. By the same example, when the ADC detects -2.8 volts, it'll record $(-2.8 + 3.7)/(7.4) \times 256 = 31 = 0 \times 1F = 000111111$.



The ADC sends these binary values to the DSP, which collects groups of them over a period of time for demodulation and audio processing, including noise detection and filtering. It also uses each group to calculate the FFT (Fast Fourier Transform) of the signal for frequency-domain examination, to eliminate harmonics, aliases, and images. In fact, numerous calculations are made on each signal group, to more efficiently produce a much cleaner signal than could otherwise be achieved with analog components alone.

Summary

As they say in ham radio, *It's better to receive than to give*, meaning that our rig design and development time is better focused on the receiver side than on the transmitter. After all, *If you can't hear 'em, you can't work 'em* is another pithy maxim that reflects the fact that being able to receive a signal is indispensable toward making a contact. With those in mind, technology advances such as *Direct Sampling* have improved our received signal integrity by reducing the need to down-convert the incoming signal, and shifting some of the workload onto the often under-utilized DSP.

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